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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

	Application No.	Applicant(s)
	10/790,070	RHODES ET AL.
Office Action Summary	Examiner	Art Unit	
	CHRISTINE DUONG	2616	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE *3* MONTH(S) OR THIRTY (30) DAYS,
WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.
- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed
  after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133).
  Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any
  earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *22 March 2004*.
2a)☐ This action is **FINAL**.   2b)☒ This action is non-final.
3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is
     closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1,3-6 and 46-65* is/are pending in the application.
     4a) Of the above claim(s) _____ is/are withdrawn from consideration.
5)☐ Claim(s) _____ is/are allowed.
6)☒ Claim(s) *1,3-6 and 46-65* is/are rejected.
7)☐ Claim(s) _____ is/are objected to.
8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.
10)☒ The drawing(s) filed on *02 March 2004* is/are: a)☒ accepted or b)☐ objected to by the Examiner.
     Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).
     Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).
11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).
     a)☐ All   b)☐ Some * c)☐ None of:
          1.☐ Certified copies of the priority documents have been received.
          2.☐ Certified copies of the priority documents have been received in Application No. _____.
          3.☐ Copies of the certified copies of the priority documents have been received in this National Stage
              application from the International Bureau (PCT Rule 17.2(a)).
     * See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)
1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☒ Information Disclosure Statement(s) (PTO/SB/08)
      Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413)
      Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

Information Disclosure Statement

The references listed in the Information Disclosure Statement, filed on 02 March 2004, have been considered by the examiner (see attached PTO-1449 form or PTO/SB/08A and 08B forms).

Claim Objections

1.   Claim 61 is objected to because of the following informalities: it is suggested to replace "telephone can support" in lines 7 and 11 and "telephone can access" in lines 8 and 12 to --telephone supports-- and --telephone accesses--, respectively. Appropriate correction is required.

Claim Rejections - 35 USC § 102

2.   The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

A person shall be entitled to a patent unless –

(e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

3.   Claims 49-53, 55 are rejected under 35 U.S.C. 102(e) as being anticipated by Shankar et al. (US Patent No. 6,570,869 B1 hereafter Shankar).

Regarding claim 49, Shankar discloses a method of choosing a call setup in a communication network (fig. 4).

The limitation, receiving a request from a calling telephone for call setup to a called telephone ("When a user initiates a voice call, a "setup" connection request message 402 is generated and ultimately transmitted by originating node 100 to originating coding unit 110" column 13 lines 12-17).

The limitation, determining whether the calling telephone and the called telephone support compatible voice compression algorithms ("The create connection message 408 can also contain parameters that indicate the capabilities of the originating coding unit 110, for example, the encoding and compression types the originating coding unit 110 supports" column 14 lines 27-32 and "During the negotiation session 414, the originating coding unit 110 and the terminating coding unit 150 negotiate appropriate compression and decoding levels" column 14 lines 61-63).

The limitation, diverting the call to a data network if the called telephone supports a voice compression algorithm that is compatible with a voice compression algorithm of the calling telephone ("The connection message 418 contains the network address of the terminating coding unit 150 and the negotiated parameters or the common capabilities" column 15 lines 7-10 and "at this point, the voice call is active" column 15 lines 49).

Regarding claim 50, Shankar discloses everything claimed as applied above (see claim 49). In addition, Shankar discloses the limitation, determining whether the calling telephone and the called telephone have access to the same data network ("the terminating coding unit 150 establishes a bearer channel circuit 416 on the bearer packet-switching network 130. The bearer channel circuit 416 may be one-way (terminating coding unit 150 to originating coding unit 110) or two-way. If successful, the terminating

coding unit 150 responds back with a connection message 418 to terminating signaling

unit 140 over control link 154" (column 15 lines 1-7).

Regarding claim 51, Shankar discloses everything claimed as applied above (see

claim 50). In addition, Shankar discloses the limitation, the data network is the Internet

("Packet-switching network 130 can be implemented as an IP network ... One example of

packet-switching network 130 is the global packet-switching network known as the

Internet" column 4 lines 15-20).

Regarding claim 52, Shankar discloses everything claimed as applied above (see

claim 49). In addition, Shankar discloses the limitation, the data network is the Internet

("Packet-switching network 130 can be implemented as an IP network ... One example of

packet-switching network 130 is the global packet-switching network known as the

Internet" column 4 lines 15-20).

Regarding claim 53, Shankar discloses a method (fig. 4).

The limitation, initiating a call setup between a calling party's audio device and a

called party's audio device using a first path ("When a user initiates a voice call, a "setup"

connection request message 402 is generated and ultimately transmitted by originating

node 100 to originating coding unit 110" column 13 lines 12-17).

The limitation, before the call setup is completed, determining whether the called

party's audio device supports a voice compression algorithm compatible with a voice

compression algorithm supported by the calling party's audio device ("The create

connection message 408 can also contain parameters that indicate the capabilities of the

originating coding unit 110, for example, the encoding and compression types the

originating coding unit 110 supports" column 14 lines 27-32 and "During the negotiation

session 414, the originating coding unit 110 and the terminating coding unit 150 negotiate

appropriate compression and decoding levels" column 14 lines 61-63).

The limitation, based on the determination, completing the call setup using a second

path different from the first path ("The connection message 418 contains the network

address of the terminating coding unit 150 and the negotiated parameters or the common

capabilities" column 15 lines 7-10 and "at this point, the voice call is active" column 15

lines 49).

Regarding claim 55, Shankar discloses everything claimed as applied above (see

claim 53). In addition, Shankar discloses the limitation, the second path includes a data

network ("Packet-switching network 130 can be implemented as an IP network ... One

example of packet-switching network 130 is the global packet-switching network known as

the Internet" column 4 lines 15-20).

Claim Rejections - 35 USC § 103

4.   The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all

obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth
> in section 102 of this title, if the differences between the subject matter sought to be patented and the prior
> art are such that the subject matter as a whole would have been obvious at the time the invention was
> made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall
> not be negatived by the manner in which the invention was made.

5.   Claims 1, 3, 5-6, 46-48, 54, 56-64 rejected under 35 U.S.C. 103(a) as being

unpatentable over Shankar in view of Berry et al. (US Patent NO. 5,758,256 hereafter

Berry).

Regarding claim 1, Shankar discloses a method (fig. 4).

The limitation, determining whether a called party's audio device is able to support at least one voice compression algorithm supported by a calling party's audio device ("The create connection message 408 can also contain parameters that indicate the capabilities of the originating coding unit 110, for example, the encoding and compression types the originating coding unit 110 supports" column 14 lines 27-32 and "During the negotiation session 414, the originating coding unit 110 and the terminating coding unit 150 negotiate appropriate compression and decoding levels" column 14 lines 61-63).

The limitation, exchanging voice signals between said called party's audio device and said calling party's audio device via a data network, if said called party's audio device is able to support said at least one voice compression algorithm ("The connection message 418 contains the network address of the terminating coding unit 150 and the negotiated parameters or the common capabilities" column 15 lines 7-10 and "at this point, the voice call is active" column 15 lines 49).

However, Shankar fails to specifically disclose the limitation, said determining step is accomplished by exchanging messages between said called party's audio device and said calling party's audio device via a circuit switched network.

Nevertheless, Berry teaches a PSTN 26 in fig. 1.

Therefore, it would have been obvious to a person having ordinary skill in the art at the time the invention was made to exchange messages between said called party's audio device and said calling party's audio device via a circuit switched network because "the packet-switching network 130 may even be overlaid on to the PSTN" (Shankar column 4 lines 17-18).

Regarding claim 3, Shankar and Berry discloses everything claimed as applied

above (see claim 2). In addition, Shankar discloses the limitation, said messages are

modified circuit control signaling messages ("common signaling protocol XISUP is an

extension of Integrated Services Digital Network User Part (ISUP)" column 12 lines 3-5).

Regarding claim 5, Shankar and Berry discloses everything claimed as applied

above (see claim 1). In addition, Shankar discloses the limitation, each of said audio

devices is one of a wired telephone, a wireless telephone, and an Internet protocol (IP)

based computer telephone ("Originating node 100 can be implemented as a Private

Branch eXchange (PBX), a telephone switch, a "smart phone" capable of generating voice

calls, a wireless PBX, or a legacy telecommunications system. Similarly, terminating node

160 can also be a PBX, telephone switch, telephone, a wireless PBX, or legacy

telecommunications system" column 3 line 65 to column 4 line 3).

Regarding claim 6, Shankar and Berry discloses everything claimed as applied

above (see claim 1). In addition, Shankar discloses the limitation, said data network is an

Internet protocol (IP) network ("Packet-switching network 130 can be implemented as an

IP network ... One example of packet-switching network 130 is the global packet-switching

network known as the Internet" column 4 lines 15-20).

Regarding claim 46, Shankar discloses a method of setting up a voice call between

two digital wireless telephones (fig. 4).

The limitation, receiving, at a switch in a communication network, a call setup

request from a calling digital wireless phone ("When a user initiates a voice call, a "setup"

connection request message 402 is generated and ultimately transmitted by originating

node 100 to originating coding unit 110" column 13 lines 12-17).

The limitation, determining, prior to completing the call setup between the calling

digital wireless telephone and a called digital wireless telephone, whether the called

wireless digital telephone uses a voice compression algorithm that is compatible with a

voice compression algorithm used by the calling digital wireless telephone ("The create

connection message 408 can also contain parameters that indicate the capabilities of the

originating coding unit 110, for example, the encoding and compression types the

originating coding unit 110 supports" column 14 lines 27-32 and "During the negotiation

session 414, the originating coding unit 110 and the terminating coding unit 150 negotiate

appropriate compression and decoding levels" column 14 lines 61-63).

The limitation, setting up the voice call over the Internet if the called digital wireless

telephone and the calling digital wireless telephone use compatible voice compression

algorithms ("The connection message 418 contains the network address of the terminating

coding unit 150 and the negotiated parameters or the common capabilities" column 15

lines 7-10 and "at this point, the voice call is active" column 15 lines 49).

However, Shankar fails to specifically disclose digital wireless telephones and

setting up the voice call over the communication network if the called digital wireless

telephone and the calling digital wireless telephone use incompatible voice compression

algorithms.

Nevertheless, Berry teaches "the origination or destination port may be a portable

subscriber unit and usable in mobile telephone environments" (column 6 lines 30-32) and

"if the user-user information indicates that a compatible FSU does not exist at the

destination port 40, the source BSC 20 does not packetize the compressed digital signal.

Instead, the source BSC 20 uses the TRF STM to decompress the voice information for

transmission over PSTN lines" (column 6 lines 10-14).

Therefore, it would have been obvious to a person having ordinary skill in the art at

the time the invention was made to have digital wireless telephones and set up the voice

call over the communication network if the called digital wireless telephone and the calling

digital wireless telephone use incompatible voice compression algorithms because it will

"minimize voice signal degradation and delay, as well as recognize the type of destination

to which a voice signal is sent" (column 1 lines 41-43).

Regarding claim 47, Shankar and Berry discloses everything claimed as applied

above (see claim 46). However, Shankar fails to specifically disclose the limitation, the

communication network is a circuit switched network.

Nevertheless, Berry teaches a PSTN 26 in fig. 1.

Therefore, it would have been obvious to a person having ordinary skill in the art at

the time the invention was made to have the communication network be a circuit switched

network because "the packet-switching network 130 may even be overlaid on to the PSTN"

(Shankar column 4 lines 17-18).

Regarding claim 48, Shankar and Berry discloses everything claimed as applied

above (see claim 47). However, Shankar fails to specifically disclose the limitation, the

circuit switched network is a public switched telephone network.

Nevertheless, Berry teaches a PSTN 26 in fig. 1.

Therefore, it would have been obvious to a person having ordinary skill in the art at the time the invention was made to have the circuit switched network be a public switched telephone network because "the packet-switching network 130 may even be overlaid on to the PSTN" (Shankar column 4 lines 17-18).

Regarding claim 54, Shankar discloses everything claimed as applied above (see claim 53). However, Shankar fails to specifically disclose the limitation, the first path includes a public switched telephone network.

Nevertheless, Berry teaches a PSTN 26 in fig. 1.

Therefore, it would have been obvious to a person having ordinary skill in the art at the time the invention was made to have the communication network be a circuit switched network because "the packet-switching network 130 may even be overlaid on to the PSTN" (Shankar column 4 lines 17-18).

Regarding claim 56, Shankar discloses everything claimed as applied above (see claim 53). However, Shankar fails to specifically disclose the limitation, based on the determination, completing the call setup on the first path.

Nevertheless, Berry teaches "if the user-user information indicates that a compatible FSU does not exist at the destination port 40, the source BSC 20 does not packetize the compressed digital signal. Instead, the source BSC 20 uses the TRF STM to decompress the voice information for transmission over PSTN lines" (column 6 lines 10-14).

Therefore, it would have been obvious to a person having ordinary skill in the art at the time the invention was made to based on the determination, complete the call setup on

the first path because it will "minimize voice signal degradation and delay, as well as

recognize the type of destination to which a voice signal is sent" (column 1 lines 41-43).

Regarding claim 57, Shankar discloses a method (fig. 4).

The limitation, sending call setup signals via a circuit-switched network to set up a

call between a calling party's telephone and a called party's telephone ("When a user

initiates a voice call, a "setup" connection request message 402 is generated and

ultimately transmitted by originating node 100 to originating coding unit 110" column 13

lines 12-17).

The limitation, consulting information relating to compatibility of respective voice

compression algorithms supported by the calling party's telephone and the called party's

telephone ("The create connection message 408 can also contain parameters that indicate

the capabilities of the originating coding unit 110, for example, the encoding and

compression types the originating coding unit 110 supports" column 14 lines 27-32 and

"During the negotiation session 414, the originating coding unit 110 and the terminating

coding unit 150 negotiate appropriate compression and decoding levels" column 14 lines

61-63).

The limitation, based on the information, sending call setup signals via a data

network to complete the call setup ("The connection message 418 contains the network

address of the terminating coding unit 150 and the negotiated parameters or the common

capabilities" column 15 lines 7-10 and "at this point, the voice call is active" column 15

lines 49).

However, Shankar fails to specifically disclose a circuit-switched network.

Nevertheless, Berry teaches a PSTN 26 in fig. 1.

Therefore, it would have been obvious to a person having ordinary skill in the art at the time the invention was made to have a circuit-switched network because "the packet-switching network 130 may even be overlaid on to the PSTN" (Shankar column 4 lines 17-18).

Regarding claim 58, Shankar and Berry discloses everything claimed as applied above (see claim 57). In addition, Shankar discloses the limitation, the information further relates to whether both the calling party's telephone and the called party's telephone have access to the data network ("the terminating coding unit 150 establishes a bearer channel circuit 416 on the bearer packet-switching network 130. The bearer channel circuit 416 may be one-way (terminating coding unit 150 to originating coding unit 110) or two-way. If successful, the terminating coding unit 150 responds back with a connection message 418 to terminating signaling unit 140 over control link 154" (column 15 lines 1-7).

Regarding claim 59, Shankar and Berry discloses everything claimed as applied above (see claim 57). In addition, Shankar discloses the limitation, the information is included in a request issued by the calling party's telephone and in a response to the request from the called party's telephone (fig. 4 shows a call flow diagram between the originating node and terminating node).

Regarding claim 60, Shankar and Berry discloses everything claimed as applied above (see claim 57). In addition, Shankar discloses the limitation, the information is included in a network node (fig. 4 shows information originating and terminating from respective node).

Regarding claim 61, Shankar discloses a method (fig. 4).

The limitation, sending a call request message to a called digital wireless telephone via a public switched telephone network as a portion of a call setup procedure, the call request message including a list of voice compression algorithms supported by a calling digital wireless telephone ("The create connection message 408 can also contain parameters that indicate the capabilities of the originating coding unit 110, for example, the encoding and compression types the originating coding unit 110 supports" column 14 lines 27-32 and "During the negotiation session 414, the originating coding unit 110 and the terminating coding unit 150 negotiate appropriate compression and decoding levels" column 14 lines 61-63).

The limitation, receiving a response message indicating whether the called digital wireless telephone can support one of the voice compression algorithms on the list, and whether the called digital wireless telephone can access a data network also accessible to the calling digital wireless telephone ("During the negotiation session 414, the originating coding unit 110 and the terminating coding unit 150 negotiate appropriate compression and decoding levels" column 14 lines 61-63 and "the terminating coding unit 150 establishes a bearer channel circuit 416 on the bearer packet-switching network 130. The bearer channel circuit 416 may be one-way (terminating coding unit 150 to originating coding unit 110) or two-way. If successful, the terminating coding unit 150 responds back with a connection message 418 to terminating signaling unit 140 over control link 154" (column 15 lines 1-7).

The limitation, if the response message indicates that the called digital wireless telephone can support one of the voice compression algorithms on the list and that the called digital wireless telephone can access the data network, completing the call setup procedure via the data network ("The connection message 418 contains the network address of the terminating coding unit 150 and the negotiated parameters or the common capabilities" column 15 lines 7-10 and "at this point, the voice call is active" column 15 lines 49).

However, Shankar fails to specifically disclose digital wireless telephones, a public switched telephone network and if the response message indicates that the called digital wireless telephone cannot support one of the voice compression algorithms on the list or that the called digital wireless telephone cannot access the data network, completing the call setup procedure via the public switched telephone network.

Nevertheless, Berry teaches "the origination or destination port may be a portable subscriber unit and usable in mobile telephone environments" (column 6 lines 30-32), a PSTN 26 in fig. 1 and "if the user-user information indicates that a compatible FSU does not exist at the destination port 40, the source BSC 20 does not packetize the compressed digital signal. Instead, the source BSC 20 uses the TRF STM to decompress the voice information for transmission over PSTN lines" (column 6 lines 10-14).

Therefore, it would have been obvious to a person having ordinary skill in the art at the time the invention was made to have digital wireless telephones, a public switched telephone network and if the response message indicates that the called digital wireless telephone cannot support one of the voice compression algorithms on the list or that the

called digital wireless telephone cannot access the data network, completing the call setup

procedure via the public switched telephone network because it will "minimize voice signal

degradation and delay, as well as recognize the type of destination to which a voice signal

is sent" (column 1 lines 41-43).

Regarding claim 62, Shankar and Berry discloses everything claimed as applied

above (see claim 61). In addition, Shankar discloses the limitation, the data network is the

Internet ("Packet-switching network 130 can be implemented as an IP network ... One

example of packet-switching network 130 is the global packet-switching network known as

the Internet" column 4 lines 15-20).

Regarding claim 63, Shankar discloses a method (fig. 4).

The limitation, performing a phase of a call setup procedure using a communication

path that includes a mobile switching center (MSC) coupled to a data network (packet-

switching network 130) and a public switched telephone network (PSTN), the phase

including sending a call request message received from a calling digital wireless telephone

at the MSC to a called digital wireless telephone via the PSTN, the call request message

including a list of voice compression algorithms supported by the calling digital wireless

telephone ("The create connection message 408 can also contain parameters that indicate

the capabilities of the originating coding unit 110, for example, the encoding and

compression types the originating coding unit 110 supports" column 14 lines 27-32 and

"During the negotiation session 414, the originating coding unit 110 and the terminating

coding unit 150 negotiate appropriate compression and decoding levels" column 14 lines

61-63).

The limitation, determining, based on a response message from the called digital wireless telephone, whether the called digital wireless telephone supports one of the voice compression algorithms on the list and whether the called digital wireless telephone has access to the data network ("During the negotiation session 414, the originating coding unit 110 and the terminating coding unit 150 negotiate appropriate compression and decoding levels" column 14 lines 61-63 and "the terminating coding unit 150 establishes a bearer channel circuit 416 on the bearer packet-switching network 130. The bearer channel circuit 416 may be one-way (terminating coding unit 150 to originating coding unit 110) or two-way. If successful, the terminating coding unit 150 responds back with a connection message 418 to terminating signaling unit 140 over control link 154" (column 15 lines 1-7).

The limitation, if the called digital wireless telephone supports one of the voice compression algorithms on the list and has access to the data network, completing the call setup procedure via the data network ("The connection message 418 contains the network address of the terminating coding unit 150 and the negotiated parameters or the common capabilities" column 15 lines 7-10 and "at this point, the voice call is active" column 15 lines 49).

However, Shankar fails to specifically disclose a MSC coupled to a PSTN, digital wireless telephone, sending a call request message received from a calling digital wireless telephone at the MSC to a called digital wireless telephone via the PSTN and if the called digital wireless telephone does not support one of the voice compression algorithms on the list or does not have access to the data network, completing the call setup procedure via the PSTN.

Nevertheless, Berry teaches a switch 25 and PSTN 26 in fig. 1 where "the switch is preferably a mobile switching center (MSC) with user-user signaling capability" (column 2 lines 65-67), "the origination or destination port may be a portable subscriber unit and usable in mobile telephone environments" (column 6 lines 30-32), "when a telephone call on the fixed wireless cellular system is initiated, the source BSC 20 exchanges call setup information with the destination BSC 30" (column 4 lines 24-27), and "if the user-user information indicates that a compatible FSU does not exist at the destination port 40, the source BSC 20 does not packetize the compressed digital signal. Instead, the source BSC 20 uses the TRF STM to decompress the voice information for transmission over PSTN lines" (column 6 lines 10-14).

Therefore, it would have been obvious to a person having ordinary skill in the art at the time the invention was made to have a MSC coupled to a PSTN, digital wireless telephone, send a call request message received from a calling digital wireless telephone at the MSC to a called digital wireless telephone via the PSTN and if the called digital wireless telephone does not support one of the voice compression algorithms on the list or does not have access to the data network, complete the call setup procedure via the PSTN because it will "minimize voice signal degradation and delay, as well as recognize the type of destination to which a voice signal is sent" (column 1 lines 41-43).

Regarding claim 64, Shankar and Berry discloses everything claimed as applied above (see claim 63). In addition, Shankar discloses the limitation, the data network is the Internet ("Packet-switching network 130 can be implemented as an IP network ... One

example of packet-switching network 130 is the global packet-switching network known as

the Internet" column 4 lines 15-20).

6.   Claim 65 is rejected under 35 U.S.C. 103(a) as being unpatentable over Shankar in

view of Farris (US Patent No. 6,064,653).

It is noted, with respect to claim 65, that the language used by Applicant merely

suggests or makes optional those features described as "adapted to"; such language does

not require steps to be performed nor limits the claim to a particular structure. It has been

held that the recitation that an element is "adapted to" perform a function is not a positive

limitation but only requires the ability to so perform. It does not constitute a limitation in

any patentable sense. In re Hutchinson, 69 USPQ 138.

Regarding claim 65, Shankar discloses a method for diverting an Integrated

Services Digital Network User Part (ISUP) network talkpath to a data network talkpath (fig.

4).

The limitation, determining, whether a called party's telephone is adapted to

exchange voice signals via a same data network to which a calling party's telephone is

adapted to exchange voice signals ("The create connection message 408 can also contain

parameters that indicate the capabilities of the originating coding unit 110, for example, the

encoding and compression types the originating coding unit 110 supports" column 14 lines

27-32 and "the terminating coding unit 150 establishes a bearer channel circuit 416 on the

bearer packet-switching network 130. The bearer channel circuit 416 may be one-way

(terminating coding unit 150 to originating coding unit 110) or two-way. If successful, the

terminating coding unit 150 responds back with a connection message 418 to terminating

signaling unit 140 over control link 154" (column 15 lines 1-7).

The limitation, establishing the data network talkpath using resources associated

with said same data network if said called party's telephone is adapted to exchange voice

signals via said same data network ("During the negotiation session 414, the originating

coding unit 110 and the terminating coding unit 150 negotiate appropriate compression

and decoding levels" column 14 lines 61-63).

The limitation, exchanging voice signals between said called party's telephone and

said calling party's telephone using the data network talkpath ("The connection message

418 contains the network address of the terminating coding unit 150 and the negotiated

parameters or the common capabilities" column 15 lines 7-10 and "at this point, the voice

call is active" column 15 lines 49).

However, Shankar fails to specifically disclose using an ISUP signaling path and

ISUP signaling path being established during a process of establishing the ISUP network

talkpath.

Nevertheless, Farris teaches "the control part of SS7 protocol is known as

Integrated Services Digital Network User Part (ISUP). ISUP determines the procedures for

setting up, coordinating, and taking down trunk calls on the SS7 network" (column 6 lines

55-59).

Therefore, it would have been obvious to a person having ordinary skill in the art at

the time the invention was made to use an ISUP signaling path and ISUP signaling path

being established during a process of establishing the ISUP network talkpath because

"signaling between switching offices is required for transmitting routing and destination information, for transmitting alerting messages such as to indicate the arrival of an incoming call, and for transmitting supervisor information, e.g. relating to line status" (column 6 lines 60-64).

7.   Claim 4 is rejected under 35 U.S.C. 103(a) as being unpatentable over Shankar and Berry further in view of Wilson (US Patent No. 6,169,734 B1).

Regarding claim 4, Shankar and Berry discloses everything claimed as applied above (see claim 1). However, Shankar and Berry fails to specifically disclose compressing said voice signals using said at least one voice compression algorithm at one of said called party's audio device and said calling party's audio device, sending said compressed voice signals to one other of said called party's audio device and said calling party's audio device via said data network and decompressing said compressed voice signals using said at least one voice compression algorithm at said one other of said called party's audio device and said calling party's audio device.

Nevertheless, Wilson teaches "allow users to receive and transmit compressed Internet voice messages across the Internet. Typically, a user at one end of the connection speaks into a microphone attached to a Personal Computer ("PC"). The microphone carries the audio voice signal to a processor board in the PC which digitizes the signal and creates a digital voice file. The voice file is then typically compressed and transferred to a selected recipient at a distant point on the Internet. Once received, the voice file is decompressed and converted via digital signal processing to an audible signal intelligible to the human ear" (column 1 lines 32-42).

Therefore, it would have been obvious to a person having ordinary skill in the art at the time the invention was made to compress said voice signals using said at least one voice compression algorithm at one of said called party's audio device and said calling party's audio device, send said compressed voice signals to one other of said called party's audio device and said calling party's audio device via said data network and decompress said compressed voice signals using said at least one voice compression algorithm at said one other of said called party's audio device and said calling party's audio device because "the device can combine the simplicity of operation of the POTS with low cost audio access to the Internet" (column 1 lines 61-62).

Citations of Pertinent Prior Art

8.   The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

Ramamurthy (US Patent No. 6,304,565 B1) discloses when a calling party calls a called party on the public-switched telephone network (PSTN), a database common to both the PSTN and an Internet Protocol (IP) network, is accessed to determine whether the called telephone line is currently active on the IP network.

Bartholomew et al. (US Patent No. 6,122,255) discloses a communication network system comprising telephone networks concatenated with a packet switched data internetwork such as the Internet.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to CHRISTINE DUONG whose telephone number is

(571)270-1664. The examiner can normally be reached on Monday - Friday: 830 AM-6

PM EST.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's

supervisor, Seema Rao can be reached on (571) 272-3174. The fax phone number for the

organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent

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applications may be obtained from either Private PAIR or Public PAIR. Status information

for unpublished applications is available through Private PAIR only. For more information

about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on

access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-

217-9197 (toll-free). If you would like assistance from a USPTO Customer Service

Representative or access to the automated information system, call 800-786-9199 (IN

USA OR CANADA) or 571-272-1000.

/Christine Duong/
Examiner, Art Unit 2616
07/31/2008

                                        /Kevin C. Harper/
                                        Primary Examiner, Art Unit 2616